

ESD-TDR-64-110

TM-03926/0000/00/0/00
(73)

THE VOCODER AND OTHER VOICE (DIGITIZING) DEVICES

TECHNICAL DOCUMENTARY REPORT NO. ESD-TDR-64-110

ESTI PROCESSED DDC TAB PROJ OFFICER ACCESSION MASTER FILE _____

MAY 1964

P. G. Edwards

ESD RECORD COPYRETURN TO
SCIENTIFIC & TECHNICAL INFORMATION DIVISION
(ESTI), BUILDING 1211

DATE _____

COPY NR. _____ OF _____ COPIES

ESTI CONTROL NR. *A6-40287*CY NR. *1* OF *1* CYS

Prepared for

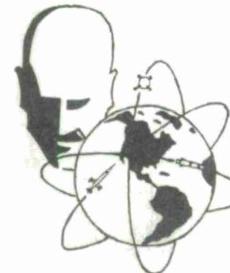
DEPUTY FOR COMMUNICATION SYSTEMS

ELECTRONIC SYSTEMS DIVISION

AIR FORCE SYSTEMS COMMAND

UNITED STATES AIR FORCE

L. G. Hanscom Field, Bedford, Massachusetts



Project 480

Prepared by

THE MITRE CORPORATION

Bedford, Massachusetts

Contract AF19(628)-2390

AD0601144

Copies available at Office of Technical Services,
Department of Commerce.

Qualified requesters may obtain copies from DDC.
Orders will be expedited if placed through the librarian
or other person designated to request documents
from DDC.

When US Government drawings, specifications, or
other data are used for any purpose other than a
definitely related government procurement operation,
the government thereby incurs no responsibility
nor any obligation whatsoever; and the fact that the
government may have formulated, furnished, or in
any way supplied the said drawings, specifications,
or other data is not to be regarded by implication
or otherwise, as in any manner licensing the holder
or any other person or corporation, or conveying
any rights or permission to manufacture, use, or sell
any patented invention that may in any way be related
thereto.

Do not return this copy. Retain or destroy.

ESD-TDR-64-110

TM-03926/0000/00/0/00

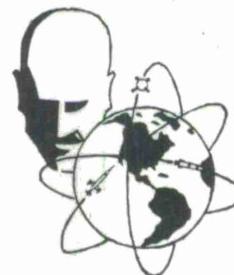
THE VOCODER AND OTHER VOICE (DIGITIZING) DEVICES

TECHNICAL DOCUMENTARY REPORT NO. ESD-TDR-64-110

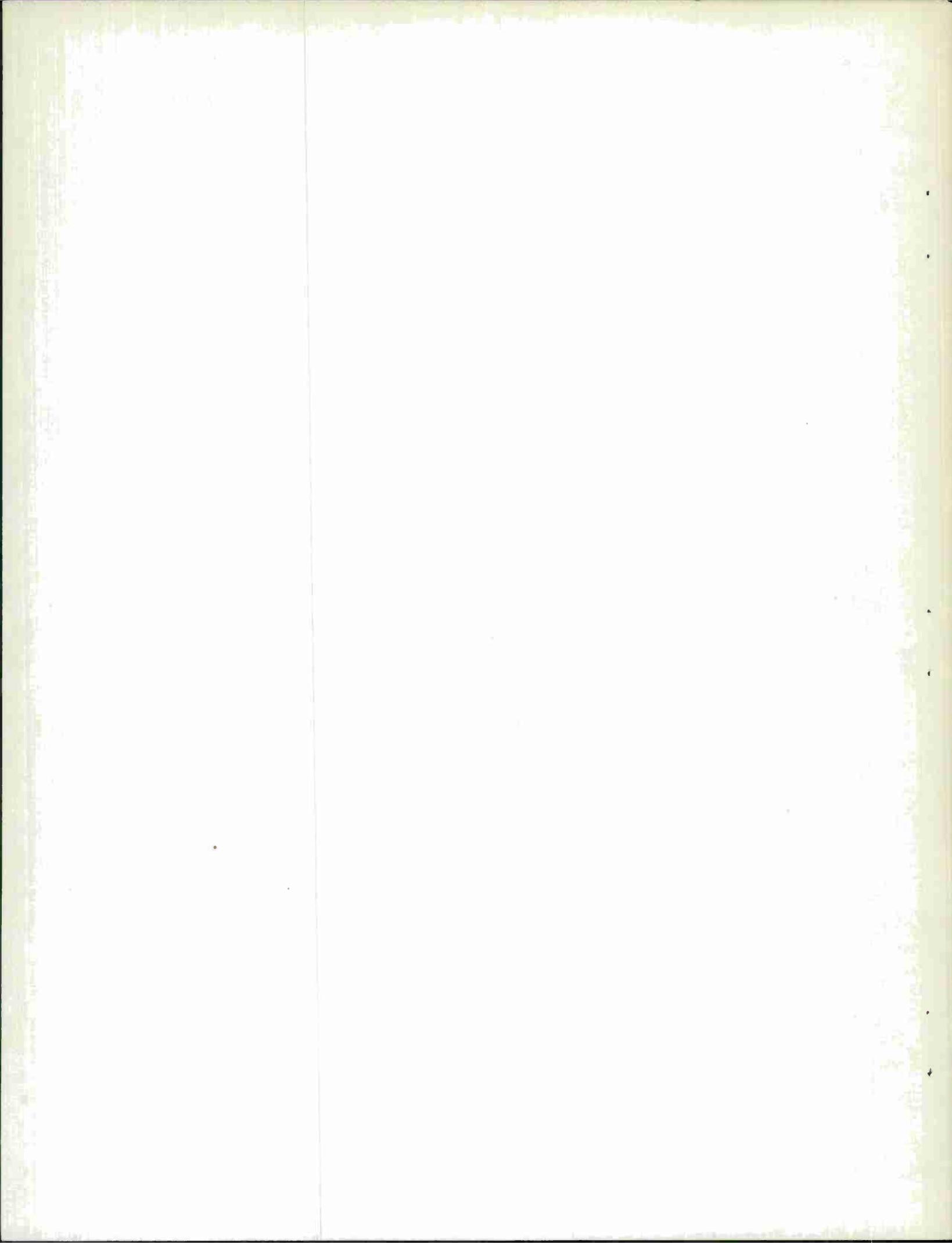
MAY 1964

P. G. Edwards

Prepared for
DEPUTY FOR COMMUNICATION SYSTEMS
ELECTRONIC SYSTEMS DIVISION
AIR FORCE SYSTEMS COMMAND
UNITED STATES AIR FORCE
L. G. Hanscom Field, Bedford, Massachusetts

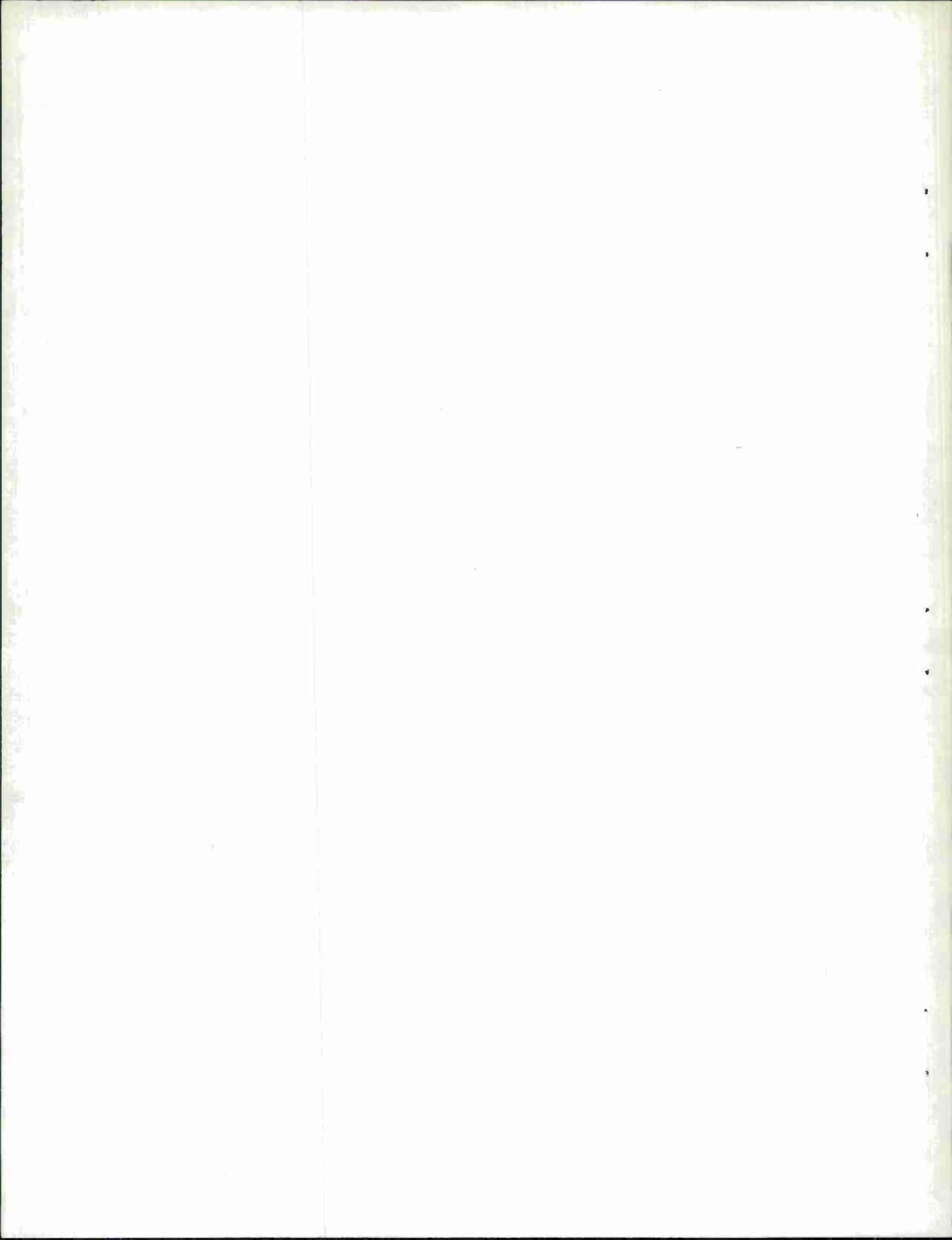


Project 480
Prepared by
THE MITRE CORPORATION
Bedford, Massachusetts
Contract AF19(628)-2390



FOREWORD

Thanks are extended to Dave Dettinger for suggesting the need for a broad, philosophical treatment of the VOCODER; to John Clapper for generating the idea of time-sharing; and to both for their constructive criticism. Much of the bibliography is drawn from papers collected by John Clapper.



THE VOCODER AND OTHER VOICE (DIGITIZING) DEVICES

ABSTRACT

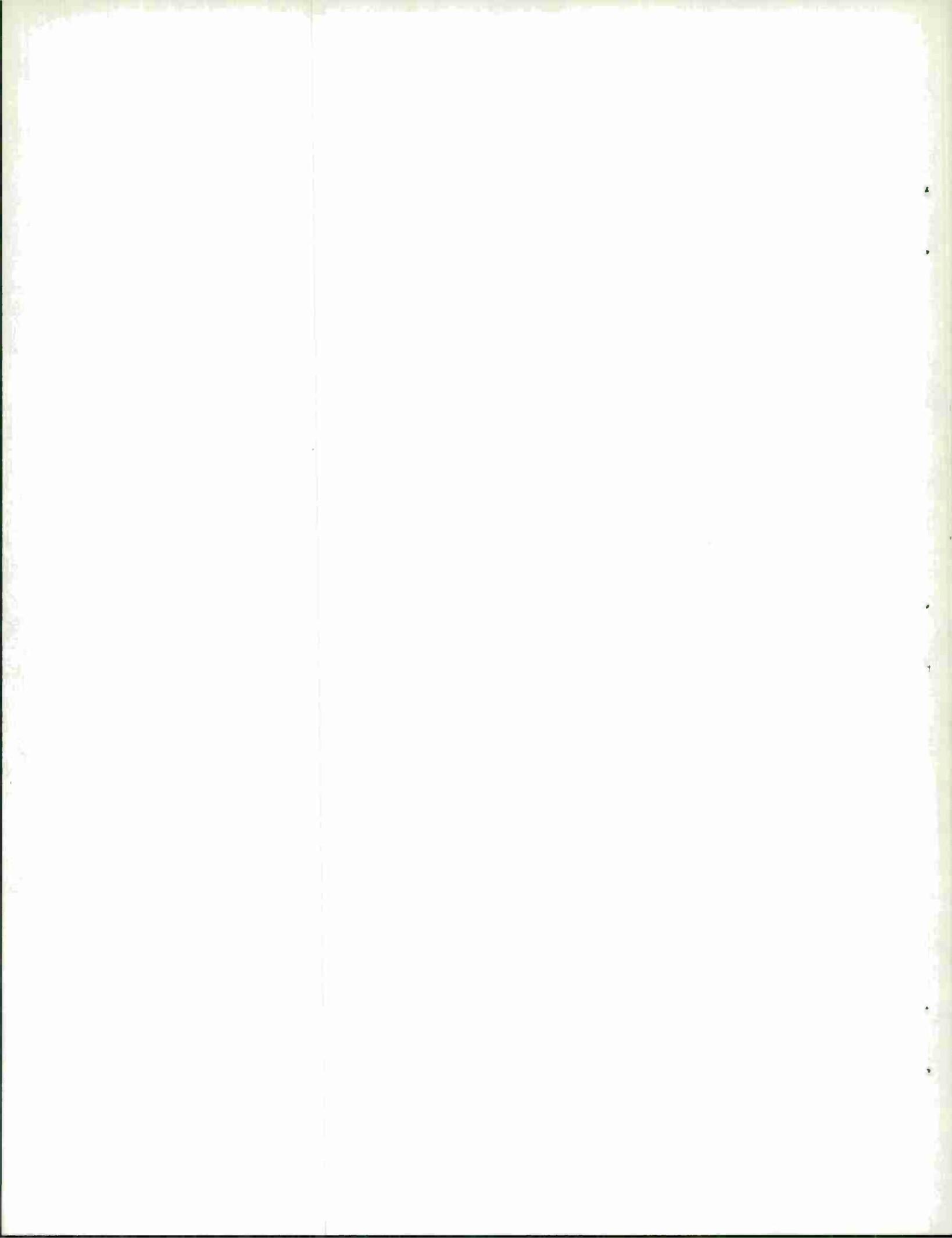
Consideration is given to the progress made in methods of voice digitizing, and, in particular, to VOCODERs as a means of providing acceptable, secure voice transmission with a minimum expenditure of bandwidth. The present situation and future trend toward this objective are discussed.

REVIEW AND APPROVAL

Publication of this technical documentary report does not constitute Air Force approval of the report's findings or conclusions. It is published only for the exchange and stimulation of ideas.



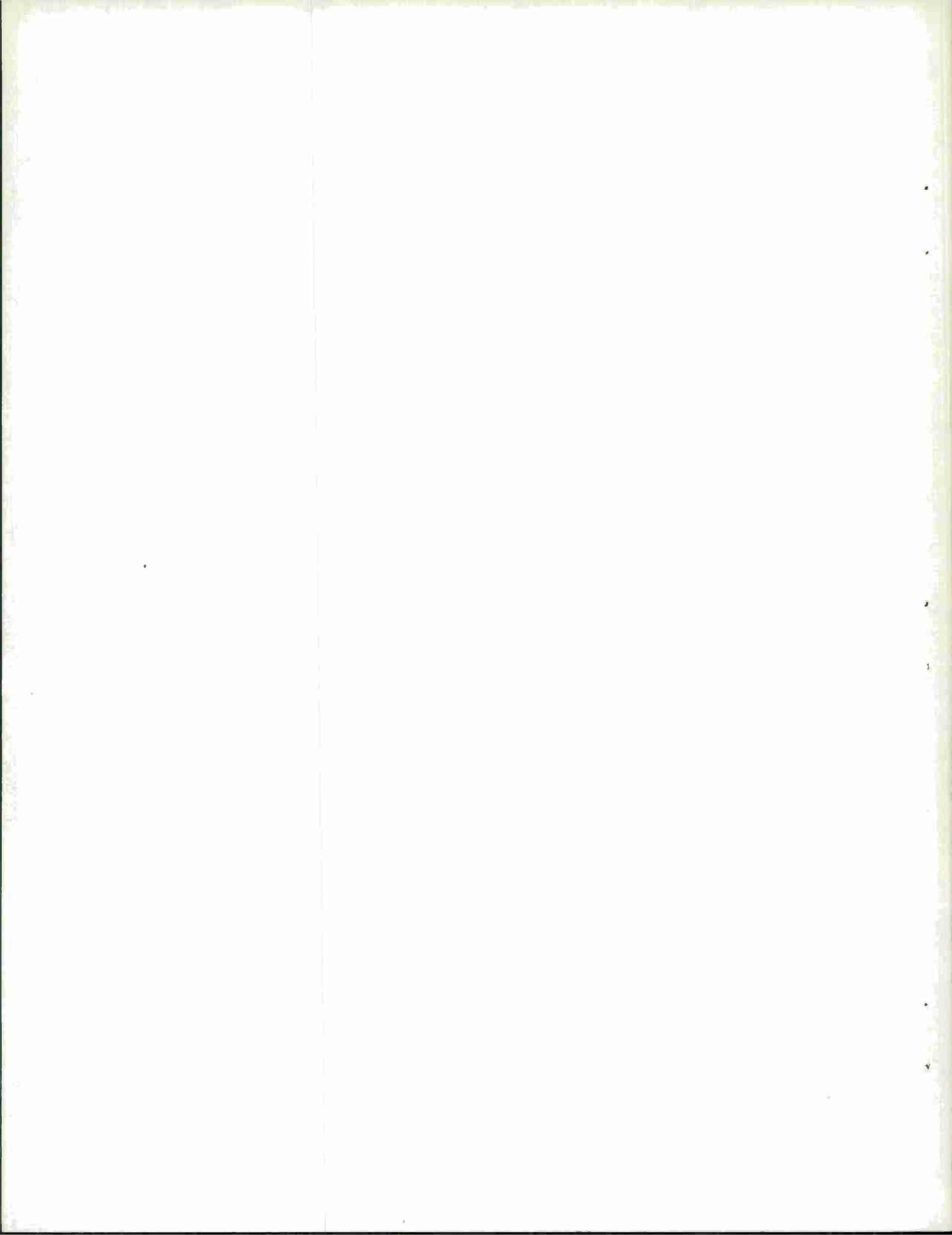
DONALD W. ROBERTS
Colonel, USAF
Deputy for Communication Systems



THE VOCODER - AND OTHER VOICE (DIGITIZING) DEVICES
SECTION I

INTRODUCTION

The matter of compressing speech in order to save transmission line bandwidth has received considerable attention over the years, as evidenced by the attached chronological bibliography covering almost a hundred references. The accompanying plot of distribution of these papers with time shows a strong surge of activity beginning in 1956, which has not yet subsided. Originally, the purpose was to transmit speech in a fraction of the original frequency band. More recently, the objective has been to minimize the bandwidth necessary to accommodate digitized speech with acceptable quality. This accomplishment may require more than the original bandwidth. It is the purpose of this report to review briefly the history of VOCODERs and similar devices, and to indicate, insofar as practicable, the probable direction of future work to meet the military needs.



SECTION II

GENERAL

The basic objective of any voice-operated device must be examined with a view to answering the question: "What can we give up that we need less than the improvement which can be thus accomplished?" The obvious redundancy in human speech has generated a number of approaches that take advantage of this redundancy in conserving the frequency bandwidth required to transmit the included information. New approaches have apparently not yet been exhausted. Bandwidth reductions of as much as 10:1 have been accomplished, but with rather severe degradation. What has been lost that is desirable? Principally, it is "quality" or "recognizability." Some of the proposed arrangements have also suffered noticeably in loss of intelligibility, but, with proper design, this kind of loss should be limited to those systems which involve drastic reductions in bandwidth.

There seems to be a general principle which says that for a given bandwidth there is a maximum quality which cannot be exceeded, but which, because of inadequate design treatment, is not necessarily achieved. List most broad principles, this one is hedged with many possible arguments. One of the biggest deterrents to proving or disproving the stated principle is that, in spite of all of the effort which has been expended, there is still no effective way of measuring quality. There are numerous indicators of quality, but, in the final analysis, the only real indication is the subjective comments of individual listeners, and these are loaded with prejudices and biases. Military users of early designs of VOCODERS have tended to be highly critical of them. A number of cases have been reported of flat refusal to use them in real-life critical situations.

While it is not the particular subject of discussion here, a similar problem of using redundancy to improve frequency bandwidth exists in the case of television.

An important step in this field has been, of course, the matter of obtaining color transmission in the bandwidth of black-and-white, in which the loss of line definition incurred by including more information in a given band has effectively been counteracted by the extra definition provided by the subjective effect of color.

SECTION III

BACKGROUND

Work in the middle 1930's, indicated by a peak in papers at 1939-1940 (see Fig. I-1), resulted in the development and demonstration of the VOCODER (Voice Coder), which, in one form, was exhibited at the 1939 New York World's Fair as PEDRO THE VODER.

In the VOCODER, the speech was analyzed and the information transmitted over a number (10) of telegraph-type (narrow) channels to the distant end, where the speech was reproduced synthetically from "buzz" and "hiss" generators which were pulsed at the frequencies of the original samples. The total bandwidth required on the transmission line was only 300 cycles. In the VODER, a human operator, with controls somewhat resembling those of a pipe-organ, created synthetic speech by manipulating these controls. The intelligibility of these devices was reasonably good, but the quality of the reproduced speech, as might be expected, was quite mechanical. Subsequent work involved increasing the number of control channels, with some improvement, but the device was still almost devoid of providing voice recognition, and there was considerable difficulty in heterosexual voice discrimination.

Another device of later origin was known as AUDREY (Automatic Digit Recognizer) and was designed primarily for the recognition of spoken digits. However, the principle of recognizing and simulating speech patterns or "phonemes" was built into a VOCODER-type arrangement. In comparing patterns and "voting" on similarities of observed and stored information, AUDREY employed a computer. There are many ways in which computers might be used to simulate speech, but all are subject to an important factor which is different from VOCODER simulation. This is the matter of amplitude or power variation. In the VOCODER, the several channels are used to describe the frequency

pattern. The amplitude variation with time of each frequency can be transmitted over the channel carrying the frequency information. With a computer involving pattern recognition of some kind, the number of patterns is greatly increased by the need to include amplitude as a part of, or as an adjunct to, the pattern.

Of about the same vintage as AUDREY was still another speech compression system known as VOBANC (Voice Band Compressor). This principle assumed that the "formants" of speech were divided into high, medium, and low bands. Frequencies were identified in each of these bands, and frequency dividers halved the frequencies transmitted on the line to yield an overall reduction in line bandwidth of 2:1.

The channel VOCODER, developed by H. W. Dudley at the Bell Telephone Laboratories before World War II, used a set of contiguous band-pass filters of about 20 cycles each, with rectifiers and low-pass filters to produce values of the short-time frequency spectrum at a series of discrete frequency points. His VOCODER recognized and differentiated between speech sounds which were either "voiced," (as a vowel) or "non-voiced" (as a sibilant). It also contained a "pitch extractor" which developed a pulsed voltage proportional to the fundamental vocal frequencies of the "voiced" sounds. At the receiving end, each channel was modulated with a buzz generator and a hiss generator pulsed in accordance with the information signaled from the "voiced" or "non-voiced" characteristics at the time and at the frequency of the "pitch extractor" signal. The filter outputs were summed, and the spectrum of speech was thus reconstructed, or synthesized. Dudley's channel VOCODER was deficient in quality.

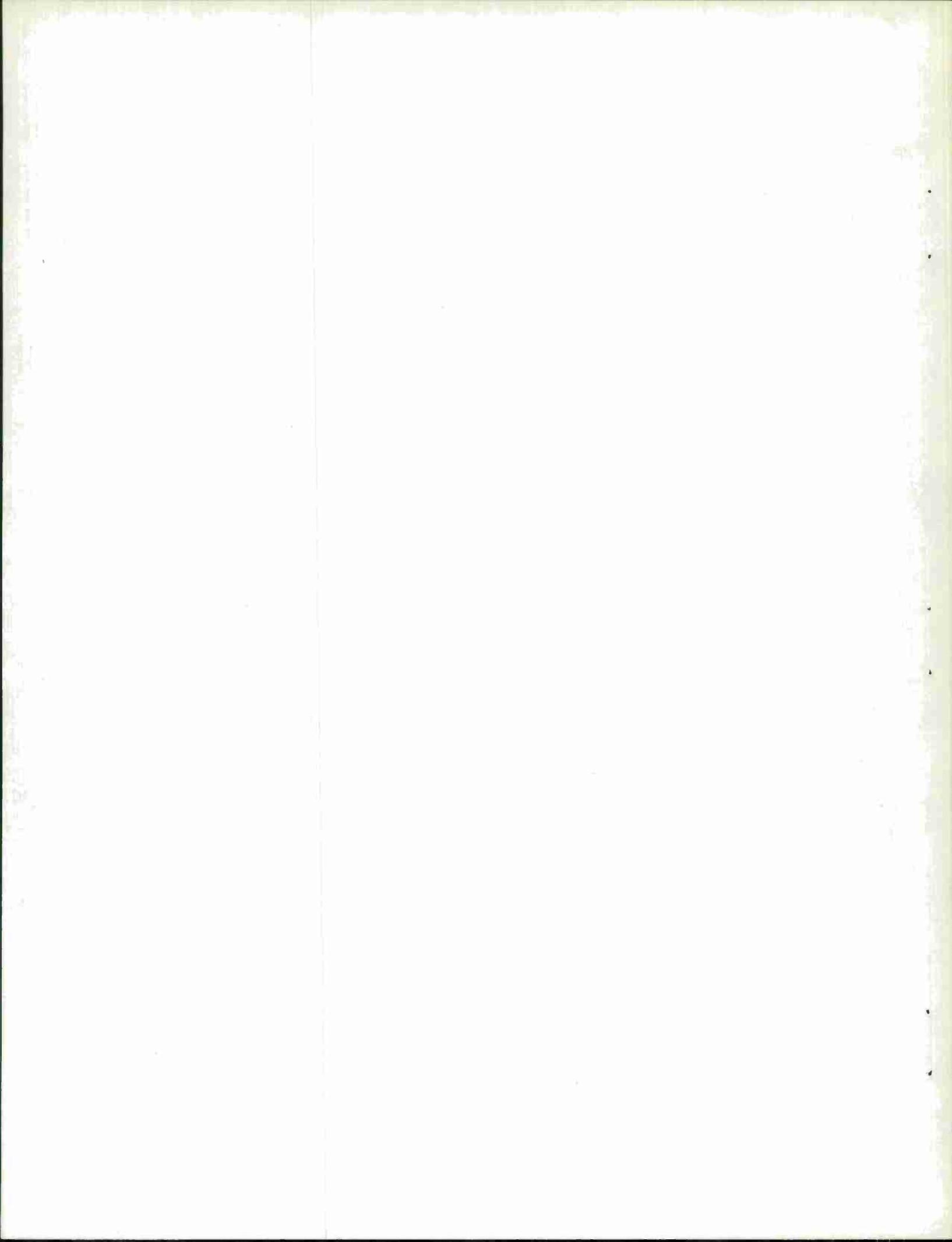
More recently, the Semi-VOCODER or VEV (Voice-Energized VOCODER) has emerged to overcome some of the disadvantages of the channel VOCODER. The lower voice frequencies from a few hundred cycles to about 800 to 1000 cycles are transmitted in analog (original) form. The channel VOCODER

principle is applied to voice frequencies in the 1000 to 3500 cycle range, it being common to use about eighteen or twenty 25 cycle channels. As compared with the channel VOCODER, this method improves naturalness and increases overall word intelligibility. It gives a bandwidth reduction of about 2:1. It is not directly applicable to existing security systems, since these require complete digitizing, and by the time the analog portion is digitized on a high-quality basis, considerably more than a voice band may be required.

A few more words are in order in regard to the amount of bandwidth required for high quality voice digitizing. The T1 carrier system is a pulse code system involving 2^7 amplitude levels, which, with two samples per cycle of maximum voice frequency, gives 56 Kc of bandwidth per voice channel:

$$7 \times 8000 = 56,000.$$

The quality of this system is equivalent to that of a direct line. How much this requirement could be reduced on a practical basis is problematic. It is interesting to note that the VOCODER can, in one sense, improve quality of transmission, since it can effectively "transmit" higher and/or lower frequencies than the connecting circuit is capable of transmitting without the VOCODER.



SECTION IV

WHAT IS NEEDED AND WHY

A particular need exist for speech security with the maximum economy reasonably attainable from the general objective for frequency conservation for two important reasons:

- (1) The application of security methods requires digitizing of speech.
- (2) A requirement exists for voice recognition, as well as intelligibility.

Item 1 is important because the better VOCODERs designed for use over a 4-Kc voice channel transmit the lower frequencies of the human voice without compression in the original analog form, while VOCODERs employing complete digitizing require higher bit rates and greater bandwidths to give satisfactory quality.

Item 2 is important because if voices are not recognizable, the security system will not be used. Those requiring security are likely to be at such levels in administrative or military hierarchies that it is not feasible to change their attitude by direction. It is the writer's conviction that, even with the existence and use of a highly secure system over a data channel, final recognition of the "go-word" would depend on the recipient asking via whatever "regular" telephone (unsecure) channel he is accustomed to use: "Do you really mean that, Charlie?" and failing to receive a satisfactory answer in kind, would feel thoroughly insecure. This is in spite of the obvious fact that "spoofing" by voice imitation is an easy art.

The real need, then, is not to determine how to put a satisfactory security system into a 4-Kc voice band. This is a desirable objective, but one which is

unlikely to be achieved. Rather, the need is to determine the frequency band which is required for a satisfactory security system, using the present art. The answer will be intermediate between a voice band and that required to digitize voice on a high-quality basis, but it may be expedient to provide the latter band, on the basis that the added cost of the frequency spectrum does not counterbalance the considerable cost of the band reduction equipments, particularly in view of the attendant loss in performance.

SECTION V

THE PRESENT SITUATION

One presently available high-quality wideband keystream unit employs 50-kilobit line bandwidth. Presently, the interconnection of these units over long distances is quite limited. The 50-kilobit signal is not accommodated by the 48-Kc group bandwidth in common use. There exists a specially designed modem unit to fit the 50-kilobit signal into more than one group band, but these have not been used except for special applications. One of the proposed security systems under development assumes the use of 38.4 kilobits for the coded information channel, so that this channel, plus signaling, could easily be included in a standard 48-Kc group band.

Secure transmission over longer distances is dependent principally upon a "standard" VOCODER which digitizes a voice band for transmission on the line at 2400 bits per second. This VOCODER output can be encoded by a suitable keystream generator. Neither this VOCODER, nor other arrangements for encoding voice for transmission in a voice band, is good enough from a quality standpoint to be widely accepted. (This standard VOCODER is presently being modified to provide 9600 VEV performance as well.) Another problem is the transmission of 2400 bits over existing commercial telephone lines without special equalization. Still another problem is the need for low voice frequencies (70 cycles up) as an input to the VOCODER. The low-frequency requirement makes necessary the use of a moving coil microphone in the associated subset, since standard desk top instruments in wide commercial use employ carbon microphones which cut off sharply below 300 cycles. It is also necessary to use connecting lines or loops capable of passing low frequencies to the VOCODER.

Other problems in connection with VOCODERS and their associated modem equipment are concerned with synchronizing and conferencing arrangements.

Conferencing is a problem because, by its very nature, a VOCODER cannot accommodate more than one talker input at a time, nor can it satisfactorily recognize speech in the presence of a substantial amount of noise. For conferencing, it is necessary to arrange for talking on a "directed" basis, where conferees are designated to talk in some organized sequence. Each sequence involves resynchronization of the network. This places a burden on the synchronizing arrangements as to time required to synchronize or restore in case of dropout.

SECTION VI

PRESENT AND PROBABLE FUTURE TRENDS

A variety of developers are pursuing a variety of paths toward improved VOCODER performance, both as to quality and intelligibility. A number of surveys has been made, rating these devices as to performance and required bandwidth. Indications are that this kind of effort will continue for some time. It is important to note that some workers are beginning to break out of the restriction to a voice channel bandwidth for transmission on the line. An important example is that of digitizing the Semi-VOCODER output, as noted above, which requires about 9600 bits for transmission.

Another important trend, entirely aside from the VOCODER problem itself, is the design of modems to squeeze more and more bits per second into a particular bandwidth. The most promising concepts are the use of phase orthogonality and multilevels. Some conservative developers feel they are close to accommodating 4800 bits in a voice channel. Whether this objective can be met or not, these techniques can improve the performance of 2400-bit transmission over voice bands which could not otherwise accommodate 2400 bits because of phase distortion. Another desirable possibility for new techniques of this kind would be the accommodation of existing 50-kilobit encoded systems in existing 48-Kc group bands, which, as noted above, is not now feasible.

A possibility worth considering is the use of existing 5-Kc radio and television sound "program" circuits for the additional quality they would afford. First, they present a wider band. Second, they transmit lower frequencies, down to 50 cps. Third, they exist in quantity, and are well distributed around the country. Finally, if the new modem techniques mentioned above are effective, it might be feasible to accommodate the 9600 bits for a digitized

Semi-VOCODER on a 5-Kc "program" circuit. Surprisingly, this approach has been given little attention to date.

The use of the Bell System T1 pulse-code carrier is proposed for two different kinds of applications. One of these involves the possibility of group encoding the approximately 1.5-megabit output of a standard 24-channel terminal. This encoded information presumably could be transmitted over the standard T1 repeatered line for whatever distance T1 is practicable. This is some 20 miles nominally. It is possible to use T1 as terminals for a radio link, in which case the distance would be limited by the capability of the radio link. The other proposed application would be the design of a new T1 terminal to operate with the standard T1 repeatered line (or radio link) to accommodate a number of 50 kilobit encoded outputs and a number of clear channels. The latter suggestion appears to be the more practical one, since it does not involve the low-frequency loss limitation imposed by using T1 terminals on a standard basis.

A factor which will have an important influence is the growing need to switch broad-band (48-Kc groups or 240-Kc supergroups). This trend arises from two sources: first, the subscriber need for 48-Kc bands, to be provided under the TELPAK arrangement; and second, the matter of broad-band restoral under emergency conditions, either manual (pushbutton) or automatically through some form of broad-band alternate routing. It is now feasible to obtain commercially, as a special arrangement, a broad-band of frequencies which can be directed through the regular switched dial network as a "piggy-back" adjunct to a clear voice channel over which the dialing is done.

An area which can take advantage of speech redundancy is that of time-sharing of facilities. This can be done in different ways. Consider first an individual talker. While perhaps 20 or more discrete VOCODER channels may be required to synthesize speech, not all of these channels are in use at one

time. It may be possible to furnish a smaller number of channels, and time-share them. It is, of course, necessary to provide means for identifying, over the connecting line, which channels are allocated to what purpose at any particular instance or period. When one considers the speed with which this identification must be made and changed, and assigns adequate channel frequency space for this control, it seems that the saving in bandwidth by time-sharing is absorbed by the need for bandwidth to do the controlling, so the net bandwidth gain is zero; in addition, there is a financial burden in furnishing the complex equipment to perform the time-sharing function. There seems to be little possible advantage in this particular concept. However, there is still the possibility of time-sharing by more than one subscriber, as TASI does on the Transatlantic Cable, except that here we are concerned with VOCODER rather than speech outputs. This concept, however, requires a number of such channels over a particular route. It seems unlikely that any economy could result even here unless, as in the case of the Transatlantic Cable, the connecting cost by normal means is high enough to absorb the cost of the time-sharing terminal equipments. The use of a wide channel, such as the 9600 bit per second channel discussed earlier, appears more promising.

While it is not directly pertinent to the subject, a few additional words can be said about television. More and more frequently the requirement for television is injected. When this requirement is for high definition color television on a secure basis, the multiplication of requirements results in a bandwidth for one channel of between 50 and 100 megacycles. This obviously encourages consideration of the possibilities of redundancy advantage. When one inquires into the reasons for high definition (1000 lines for example), it usually results from the desire to reproduce page-print or the like without error. When such material is sent, it does not have to be at a high repetition rate, so one way to reduce the bandwidth is to provide different speeds for different kinds of material.

Another method, which also is well-known, is to minimize the number of amplitude levels sent. So far there seems to be nothing in progress for television comparable to the VOCODER for voice. For certain types of transmitted material, such a method might have considerable advantage.

In a following paper it is planned to describe the different VOCODER approaches in more detail, as well as particular entities of each kind, and to compare the results of tests of several VOCODERS.



P. G. Edwards

APPENDIX I

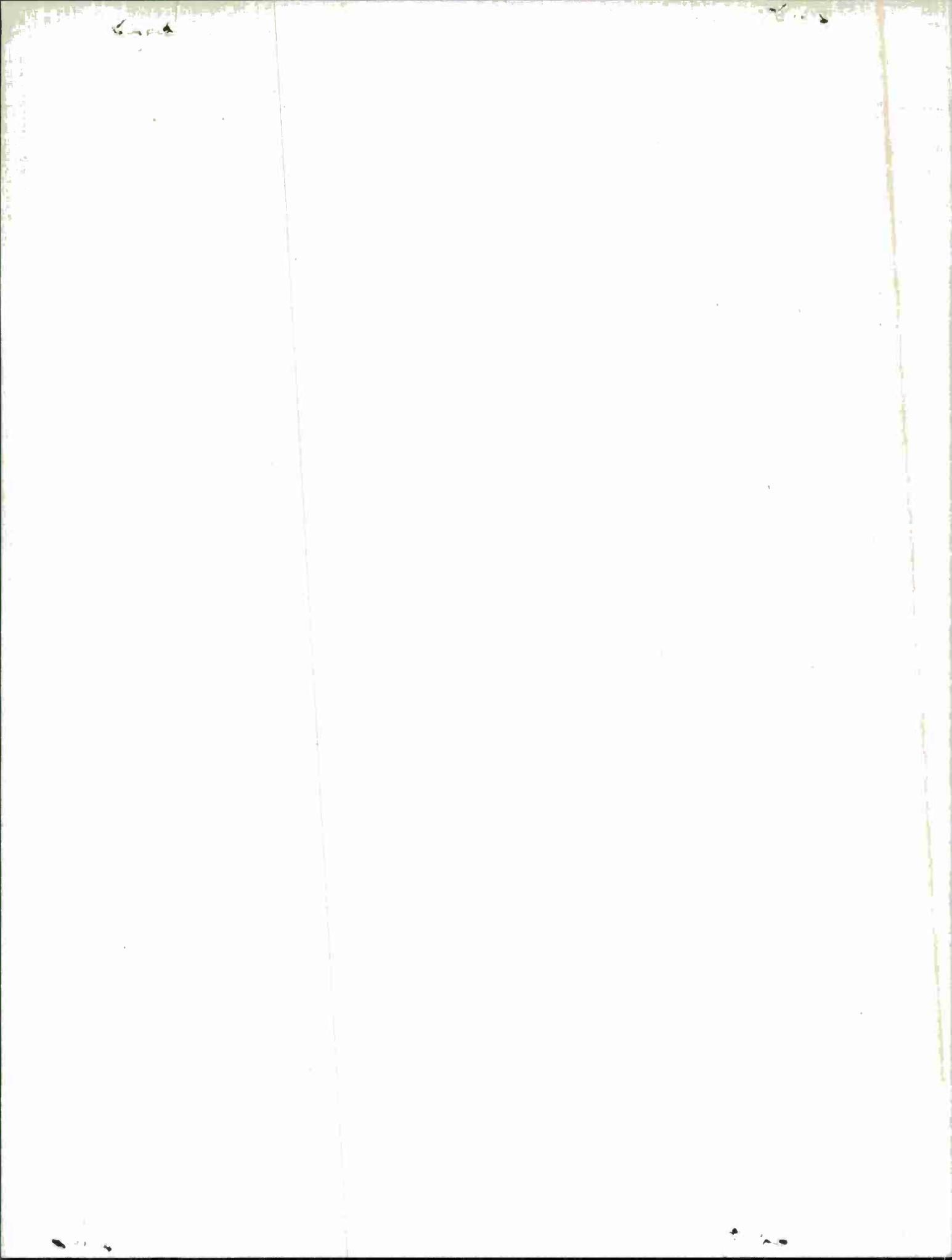
CHRONOLOGICAL BIBLIOGRAPHY
SPEECH SYNTHESIS - VOCODERS

(All References are unclassified)

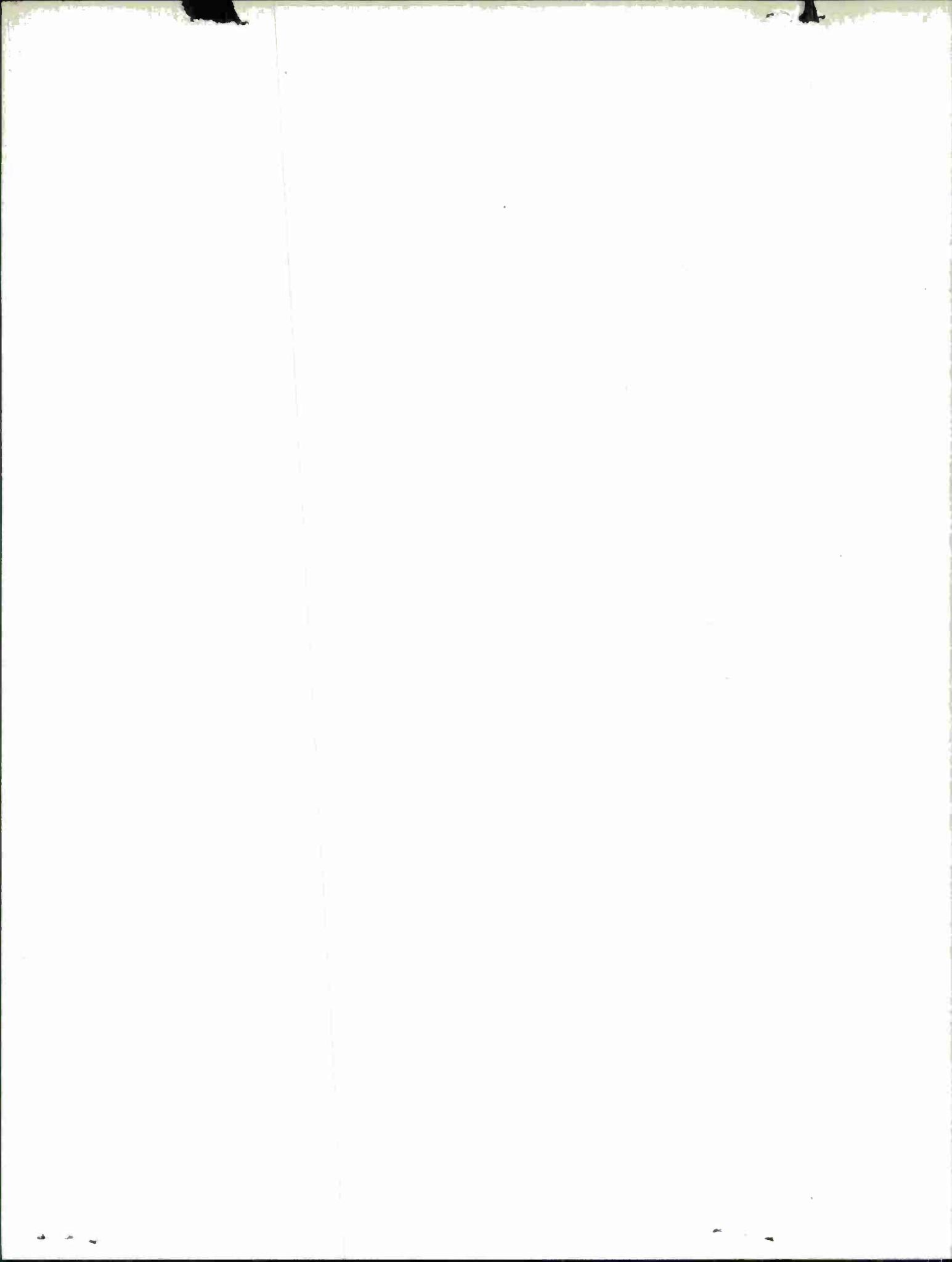
| | <u>Year</u> |
|--|--------------|
| Bell System Tech. J. <u>1</u> , No. 1 Analysis of the Energy Distribution of Speech Irving B. Crandall | 1922 |
| Bell System Tech. J. 586-626 The Sounds of Speech Irving B. Crandall | Oct. 1925 |
| Bell System Tech. J. 349-373 Some Physical Characteristics of Speech Harvey Fletcher | July 1931 |
| J. Franklin Institute <u>227</u> , 739-764 A Synthetic Speaker Dudley, Riesz, and Watkins | June 1939 |
| Proc. Natl. Acad. of Sci. <u>25</u> , 377-383 The Automatic Synthesis of Speech Homer Dudley | July 1939 |
| J. Acoust. Soc. Am. <u>11</u> , 169-177 Remaking Speech Homer Dudley | Oct. 1939 |
| Bell Lab. Record The Vocoder H. W. Dudley | 1939 |
| Bell System Tech. J. 495-515 The Carrier Nature of Speech Homer Dudley | Oct. 1940 |

| | <u>Year</u> |
|---|---------------|
| J. IEE <u>95</u> , Part III, 391-411 Analysis-Synthesis Telephony with Special Reference to the Vocoder R. J. Halsey and J. Swaffield | Sept. 1948 |
| U.S. Patent 2, 635, 146 Speech Analyzing and Synthesizing Communication System J. C. Steinberg | Dec. 1949 |
| J. Acoust. Soc. Am. <u>22</u> , No. 2 151-166 The Speaking Machine of Wolfgang Von Kempelen H. Dudley, T. H. Tarnoczy | March 1950 |
| Bell Lab. Record An Electrical Vocal System L. O. Schott | Dec. 1950 |
| J. Acoust. Soc. Am. <u>22</u> , 820-823 The Intelligibility of Amplitude-Dichotomized Time-Quantized Speech Waves J. C. R. Licklider | 1950 |
| Bell Lab. Record Vocal Gestures Gordon E. Peterson | Nov. 1951 |
| J. Acoust. Soc. Am. <u>25</u> , 832 Improvements in the Vocoder, (Abstract) R. L. Miller | July 1953 |
| J. Acoust. Soc. Am. <u>26</u> , 137 The VOBANC, A Voice Band Compressor B. P. Bogert | 1954 |
| IRE Trans. on Audio, AU-2, 7-12 Method for Time or Frequency Compression- Expansion of Speech G. Fairbanks, W. L. Everitt and R. P. Jaegar | 1954 |

| | <u>Year</u> |
|---|--------------------|
| ASTIA No. AD-119258 A Summary of VOCODER Research over the Period 1947-1955 A. C. Hales, T. R. Boxhall and A. D. S. Cullis | 1947 to 1955 |
| Bell Lab. Record Speech Bandwidth Compression W. E. Kock | March 1956 |
| J. Acoust. Soc. Am., <u>28</u> , 767-768 Abstr. Technique for Coding Speech Signals for Transmission Over a Reduced Capacity Digital Channel E. E. David, Jr. and H. S. McDonald | 1956 |
| Butterworth Scientific Publications, London, Info. Theory 231-244 New Methods of Speech Transmission P. Marcau and J. Daguet | 1956 |
| J. Acoust. Soc. Am. <u>28</u> , 1091-1098 Speech Analysis-Synthesis Scheme Using Continuous Parameters C. R. Howard | 1956 |
| J. Acoust. Soc. Am. <u>28</u> , 1099-1106 Development and Testing of a Format - Coding Speech Compression System J. L. Flanagan and A. S. House | 1956 |
| J. Acoust. Soc. Am. <u>28</u> , 586-589 Naturalness and Distortion in Speech Processing Devices E. E. David, Jr. | 1956 |
| J. Acoust. Soc. Am. <u>28</u> , 399-404 The VOBANC--A Two-to-One Speech Band Width Reduction System B. P. Bogert | 1956 |
| J. Acoust. Soc. Am. <u>29</u> , 306-310 Note on the Design of "Terminal-Analog" Speech Synthesizers J. L. Flanagan | 1956 |



| | Year |
|--|--------------|
| IRE Trans. on Circuit Theory, CT-3, 232-244 Signal Theory in Speech Transmission E. E. David, Jr. | 1956 |
| AFCRC TR-57-111, ASTIA No. AD-117290 Speech Data Reduction. Voice Communications by Means of Binary Signals at Rates under 1000 Bits per second C. P. Smith | Jan. 1957 |
| Final Report Cont. AF 33 (600) 32293, Melpar, Inc. Development of a Continuous Analysis Speech Compression System T. E. Bayston and S. J. Companella | July 1957 |
| Quarterly Progress Report, MIT Res. Lab. for Electronics The Extraction of Pitch Information from Speech A. R. Adolph | July 1957 |
| Bell Lab. Record Voice Actuated Machines--Problems and Possibilities E. E. David, Jr. | Aug. 1957 |
| U.S. Patent 2,817,711 Band Compression System C. B. H. Feldman | Dec. 1957 |
| J. Acoust. Soc. Am. <u>29</u> , 636-641 Word Intelligibility as a Function of Time Compression G. Fairbanks and F. Kidman, Jr. | 1957 |
| J. Acoust. Soc. Am. <u>29</u> , 777 Peakpicker: A Bandwidth Compression Device E. Peterson and F. S. Cooper | 1957 |
| Trans. IRE Prof. Group on Audio, Vol. AU-6 No. 5 (Sept. - Oct. 1958) 104-116 A Survey of Speech Bandwidth Compression Techniques S. J. Campanella | 1958 |



| | <u>Year</u> |
|--|---------------|
| AFCRC ERD-CRRS - TM-58-103 A Method for Speech Data Processing by Means of a Digital Computer C. P. Smith | 1958 |
| J. Acoust. Soc. Am. 30, 201-209 A Dynamic Analog Speech Synthesizer G. Rosen | 1958 |
| J. Acoust. Soc. Am. 30, 733-739 Phonetic Pattern Recognition VOCODER for Narrow-Band Speech Transmission H. W. Dudley | 1958 |
| AFCRC-TR-59-116, ASTIA No. AD-214449 Pitch Extractor for Speech Synthesis with Special Techniques for Use in Digital Bandwidth Compression Systems L. V. Kriger | March 1959 |
| Lincoln Lab. G-Report 36-41 A Method of Speech Compression R. M. Lerner | Aug. 1959 |
| Proc. of Seminar on Speech Compression and Processing AFCRC-TR-59-198 A Twelve-Channel Transistorized Recorder K. C. H. Stockhoff | Sept. 1959 |
| Analysis by Synthesis M. Halle and K. N. Stevens | |
| The SRDE Speech Bandwidth Compression Project L. G. Stead and E. T. Jones | |
| The Formoder as a Tool for Speech Studies S. W. Chang, R. E. Bach, C. R. Howard, R. Sukys | |
| An Approach to Speech Bandwidth Compression C. P. Smith | |

| | <u>Year</u> |
|---|----------------------|
| Vocoders for Military Use M. R. Schroeder | |
| AFCRC-TN-59-791, ASTIA No. AD-229416 Contract AF 19 (604)-5579, Melpar, Inc. | Oct. 1959 |
| Voice Data Processing Systems L. P. Schoene, Jr. | |
| AFCRC-TN-59-958, ASTIA No. AD-257227 The Use of Digital Computers in Speech Analysis and Synthesis C. P. Smith | Nov. 1959 |
| SM Thesis MIT Automatic Extraction of Fundamental Period in Speech Sounds H. Fujisaki | 1959 |
| Trans. Inst. Radio Engrs. IT-5129-136 Extremal Coding for Speech Transmission M. V. Mathews | 1959 |
| IRE WESCON Conv. Record Part 7, 5-17 Resonance-Vocoder and Baseband Complement Hybrid Speech Transmission J. L. Flanagan | 1959 |
| AFCRL-TN-59-958 The Use of Digital Computers in Speech Analysis and Synthesis C. P. Smith | 1959 |
| Trans. IRE Audio Group, AU-8, No. 1, 20-26 Bandwidth Compression by Means of Vocoders F. Slaymaker | Jan. Feb. 1960 |
| SM Thesis (Unpublished) MIT An Autocorrelation VOCODER T. H. Crystal | June 1960 |
| Quarterly Tech. Report No. 4 for USASRDL, Contract DA-36-039 SC 78078 Study of a Speech Compression System (Spectrum Selection) K. D. Kryter | Sept. 1960 |

| | <u>Year</u> |
|---|----------------------|
| RADC-TN-60-196 Auto-Recognition of Speech T. Marill | Oct. 1960 |
| RADC-TN-60-197, ASTIA No. AD 245025 Review of Existing Speech Compression Systems K. N. Stevens | Oct. 1960 |
| J. Acoust. Soc. Am. <u>32</u> , 15-24 Human or Machine? A Study of Voice Naturalness E. E. David, Jr., L. G. Kersta, P. D. Bricker | 1960 |
| J. Acoust. Soc. Am. <u>32</u> , 547-556 Speech Bandwidth Compression Through Spectrum Selection K. G. Kryter | 1960 |
| Furtschritte der Hochfrequenztechnik - Akademische Verlagsgesellschaft mbH Frankfurt A/M, 229-262 Systems for Speech Compression C. G. M. Fant and K. N. Stevens | 1960 |
| Acoutica <u>10</u> , 35-43 A Vocoder for Transmitting 10 Kc Speech Over a 3.5 Kc Channel E. E. David, Jr. and M. R. Schroeder | 1960 |
| Royal Inst. Tech., AFCRL 390 Contract AF 61 (052)-342 Speech Analysis and Synthesis G. Fant | Jan. 1961 |
| MIT Lincoln Lab. Quarterly Prog. Rep., Div. 5, AFCCDD-TN-61- 1006 21-23 Pitch Extraction on TX 2 Computer | March 1961 |
| J. Acoust. Soc. Am. <u>33</u> , 597-603 Perturbations in Vocal Pitch P. Lieberman | May 1961 |
| TSEC/HY-2 Channel Vocoder, Modified Report No. 1 Contract 2342 Philco Corp. ASTIA No. AD-266389 | June Aug. 1961 |

| | <u>Year</u> |
|--|---------------|
| ITTC Task C 3061-0230, Contract AF 19 (626)-5 Subscriber Activity-Measurement of Some Performance Factors of VOCODERS | Oct. 1961 |
| <u>J. Acoust. Soc. Am. 33, Abstracts</u> | |
| Analog Multiplexing of a Telephone Semi-Vocoder A. S. Howell <u>et al.</u> | Nov. 1961 |
| 1663 Design vs Performance Factors for Some Speech Compression Systems C. P. Smith | |
| 1684 New Applications of Voice Excitation Vocoders E. E. David, <u>et al.</u> | |
| 1663 Military Semi-Vocoder for Analog Telephone Transmission A. S. Howell <u>et al.</u> | |
| ----- | |
| Proc. III ICA, Stuttgart, 1959, Elsevier Pub. Co. Amsterdam, Vol. I 217-220 | 1961 |
| Auto. Extraction of the Excitation Function of Speech with Particular Reference to the Use of Correlation Methods J. S. Gill | |
| Proc. III ICA, Stuttgart, 1959, Elsevier Pub. Co. Amsterdam Vol. I 221-224 | 1961 |
| A Study of the Requirement for Excitation Control in Synthetic Speech J. S. Gill | |
| J. Acoust. Soc. Am. 33, 179-186 Pitch Synchronous Analyses of Voiced Sounds M. V. Mathews, J. E. Miller, E. E. David, Jr. | 1961 |
| RADC-TDR-62-171 or Report No. 914, Contract AF 30 (602) 2235 Bolt, Beranek & Newman, Inc. An Evaluation of Speech Compression Systems K. N. Stevens, M. H. L. Hecker and K. D. Kryter | March 1962 |

| | <u>Year</u> |
|--|---------------|
| AFCRL 62-314, Contract AF 19 (604)-5579, Final Report, Melpar, Inc. | March 1962 |
| Design and Development of a Digital Voice Data Processing System | |
| L. P. Schoene, H. A. Straight and O. C. King | |
| RADC-TDR-62-171 | March |
| An Evaluation of Speech Compression Systems | 1962 |
| Bell Lab. Record, 216 Editorial | June |
| Computer Synthesizes Human Speech | 1962 |
| AFCRL 62-394, Contract AF 19 (628)-302, Texas Inst., Inc. | June |
| SP 14-A62, Scientific Report No. 1 | 1962 |
| Full Duplex Digital VOCODER | |
| Lawrence A. Yaggi, Jr. | |
| J. Acoust. Soc. Am. <u>34</u> , No. 7, 916-921 | July |
| Computer Program for Pitch Extraction | 1962 |
| Bernard Gold | |
| Fourth Intl. Congress on Acoustics, Copenhagen, 21-28 | Aug. |
| Description of a Computer Program for Pitch Detection | 1962 |
| Bernard Gold | |
| IRE Trans., Info. Theory <u>8</u> , No. 5 | Sept. |
| Voice Excited Vocoder for Practical Speech Band | 1962 |
| Width Reduction | |
| E. E. David, Jr., M. R. Schroeder, B. F. Logan | |
| A. J. Prestigiacomo | |
| AFCRL-62-946, Contract AF 19 (604)-8828, Data Syst. Div., | July |
| Litton, Final Report | Oct. |
| Investigation of Automation of Speech Processing for | 1962 |
| Voice Commun. | |
| Geo. S. Sebestyen and David Van Meter | |
| NEREM Proc. | 1962 |
| Speech Compression Studies | |
| C. P. Smith | |

| | <u>Year</u> |
|---|--------------|
| AFCRL-62-7, Contract AF 19 (604)-8042, EPSCO, Inc. Analog/Digital Multiplex Equipment for Voice Signal Processing D. Hertz | 1962 |
| AFCRL-62-719, Contract AF 19 (604)-5579, Melpar, Inc. Speech Data Processing in Real Time H. A. Straight | 1962 |
| RADC-TDR-63-90, Contract AF 30 (602)-2235, Balt. Beranek and Newman An Evaluation of Speech Compression Techniques Karl D. Kryter and Jay H. Ball | Feb. 1963 |
| Lincoln Lab., Contract AF 19 (628)-500 Vocoded Speech Bernard Gold | May 1963 |
| J. Acoust. Soc. Am. <u>35</u> , No. 5, 730-731 Pitch Induced Spectral Distortion on Channel Vocoder Bernard Gold and Jas. Tierney | May 1963 |
| Final Report 1192-291-42, Contract DA-36-039 AMC-00047 (E) Gen. Atronics A Multilevel Narrow Band Transmission Modem R. W. Spencer and D. N. Thomson | May 1963 |
| Proc. Natl. Electron. Conf., Chicago The Present Status of Vocoder Speech Bandwidth Reduction Systems E. E. David, Jr. | Oct. 1963 |
| J. Acoust. Soc. Am. <u>35</u> , 805 (A) Voice Communications Method Using Pattern Matching for Data Compression C. P. Smith | 1963 |

I-27

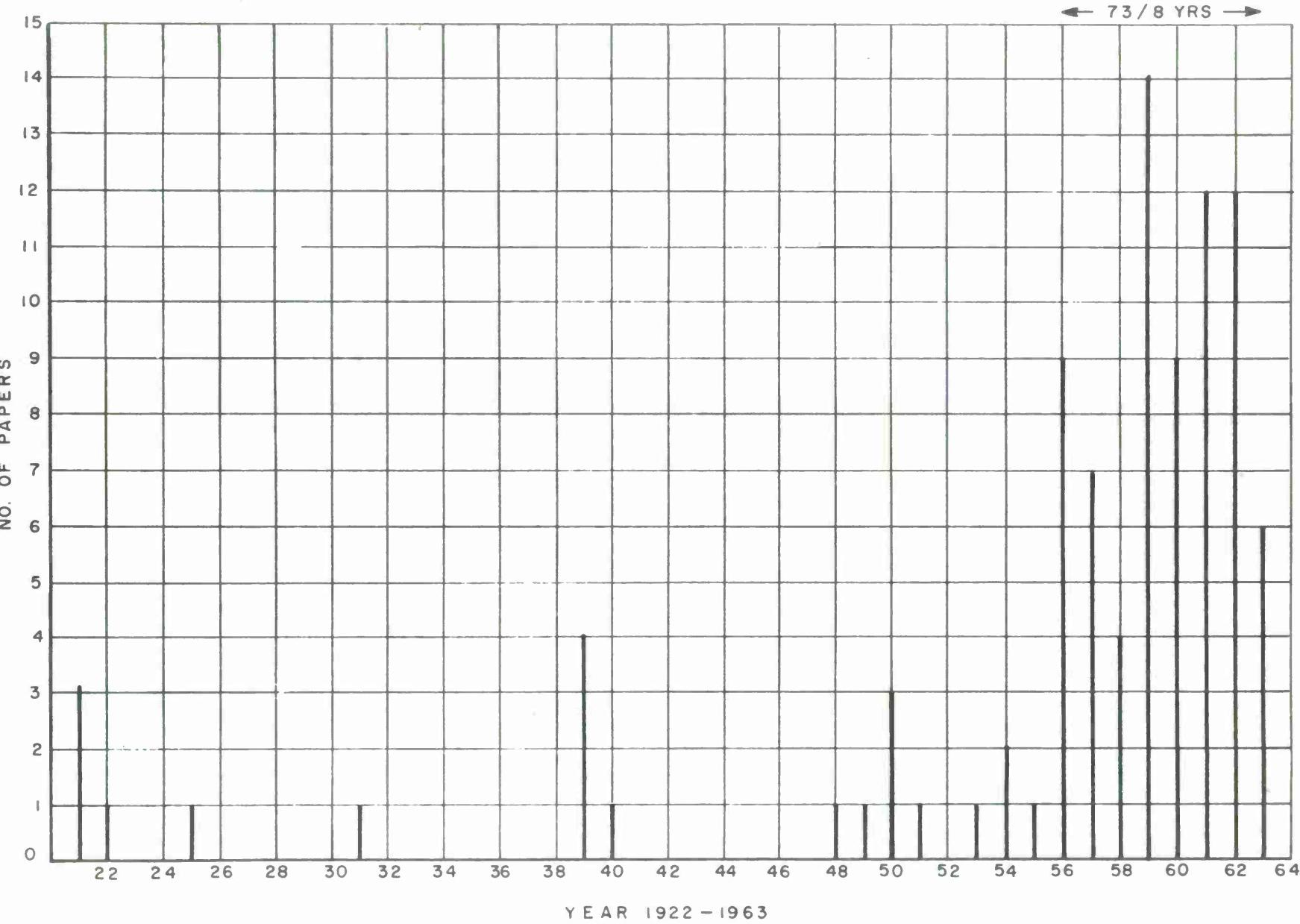


Fig. 1 SPEECH SYNTHESIS TIME DISTRIBUTION OF 91 PAPERS

